Remarks:

Reconsideration of the application is respectfully requested.

Claims 1 and 3-37 are now in the application. Claims 1, 33 and 36 have been amended.

The specification has been amended by removing therefrom a hypertext-type Internet address on page 17. The specification and claims now meet the requirements of 35 U.S.C. § 112, first and second paragraphs. Should any further objections remain, the Examiner is respectfully requested to telephone counsel so that the matter may be resolved.

The Examiner is thanked for the withdrawal of the prior Office action and the issuance of this new action. Most of the claims have now been rejected over the combined teachings of Voit et al. (US 6,157,636, "Voit") and Smyk (US 6,597,686) under 35 U.S.C. § 103.

Before delving into the details of the rejection and the combined teachings of the references, it appears that a brief review of the underlying technology would be beneficial. A proper understanding of the technical aspects forms a proper basis for the factual record, as it is paramount that the claims be read with a view to how the claims would be understood by a person of skill in the art.

We deal here with packet-switched communication. Such packet-switched communication is generally referred to as Internet protocol (IP) data communication.

Packets are collections of digital signal pulses that contain a variety of information, including address information (originating address, destination address), signaling and timing information (preamble, midamble, etc.), and useful information (data content). Data information is divided and organized into packets at the originating address (the calling user) and the packets are individually and separately transmitted to the destination address (the called user). There, the useful information is decoded from the packets, collected and assembled into the complete message and then processed as such at the destination address. Internet protocol communication has proven so efficient and successful primarily because of the packet-switched transmission protocol.

The claims of the instant application call for communication in the Internet Protocol (IP) data network. The calling user and the called user are in the IP data network and they communicate in the packet-switched protocol.

Claim 1, for example, has been further clarified to expressly state that the "destination address" lies "within the Internet Protocol data network." That is, according to claim 1, the calling user generates a data packet, the data packet is received by a network node, the network node checks information regarding the calling user and/or the services that are available to the calling user, and sends the packet on its way to a destination address in the IP data network. The path to the next node for the data packet is uniquely determined based on a "second piece of information" that is available to the network node.

The cited prior art is different. The references Voit and Smyk pertain to Internet telephony – generally referred to as VoIP (voice over Internet Protocol). The data stream in VoIP is packet-switched only intermediately. First, the voice stream (a sound file) is converted to a packet-switched stream – i.e., the sampled voice file is packaged by inserting digital equivalents into the packet segment "useful information" and as many packets as necessary to fit enough information are assembled – and the packets are sent from the "calling user" to a network node. At the network node, the packets are checked for their address and transmitted through the Internet. The address is an ITG (Internet telephony gateway), where the useful information is decoded from the packets and assembled into a (analog) data stream that is suitable for a circuit switched network (PSTN). There, the message from the calling user travels through the circuit to a destination address (the called user) which is outside of the Internet Protocol data network.

The foregoing simplified description represents the primary reference Voit. There, the calling user's voice is subjected to voice-to-IP software conversion at the computer 110. That is, the voice data stream is "packaged" into packets that are suitable for IP protocol transmission. From the computer 110, the packet-switched data is transmitted to a network node 106 (Internet, intranet), where the system looks for additional information in the directory services 114 and in a file referred to as Authentication Security Accounting 116. The network node then looks for the most advantageous ITG (Internet Telephony Gateway) that serves the <u>called user</u> and connects to one of the available ITGs 118. At the ITG, the packets are "collected," the useful information is decoded and re-assembled, and the voice data stream is

transmitted through the circuit switched network 108 to the called user at the telephone terminal 112.

Fig. 1B of Voit illustrates the system in which a "virtual telephone" (i.e., the laptop computer 110) can reach a telephone subscriber 112 through the Internet and a PSTN to which the telephone 112 is connected. The Internet and the PSTN are connected to one another through the ITG, col. 9, lines 31-39, and the request to the Directory Service 114 – based on the called number – provides the suitable ITG.

The text cited by the Examiner, namely, columns 13 and 14, relate to Fig. 6 and the VoIP objects and interface relationships which have been defined between internal ITN (=Internet Telephony Network) objects."

It is entirely clear from Voit – see especially Fig. 1B and Fig. 12 – that the address implementation and the pathway through the Internet is based exclusively on the address E.164 of the called user. That is, the ITG that is responsible for the called telephone number to form the bridge between the packet-switched network and the circuit switched network is selected based on the target E.164. No such query at the network node in Voit leads to a determination which route the packet should take within the packet-switched network.

The final feature of claim 1, namely, "uniquely determining . . . within the Internet Protocol data network," is clearly not shown or suggested by Voit. Please note the carefully inserted definition, according to which the "destination address" lies within the IP data network.

The secondary reference Smyk is quite similar to Voit, in that it also deals with VoIP and its attendant conversion across two or more networks (IP and PSTN). Smyk deals with the concept of ITG routing – see the discussion of Voit above – in the introductory text. There, the route to the ITG is defined in dependence on the <u>called party telephone number</u> as well. Col. 2, lines 19 et seq..

The Examiner made specific reference to text that pertains to Fig. 4: A calling user (telephone terminal set 402) establishes a connection to his local PSTN 410 and into an SSP (signal switching point) 408 of the circuit switched network.

The SSP 408 and a SCP 404 (service control point) are network elements of a so-called intelligent network (IN, according to the ITU-Standard Q.1200 ff). The IN is a service-oriented central system which piggy-backs onto a regular telephone network (for example an ISDN). Such intelligent networks are provided to augment regular telephone networks PSTN with intelligent network components and additional service features.

Smyk's SCP determines a route to a POP, which offers a connection to the most suitable telephone service carrier with the best performance to cost ratio. The selection and the further transmission is effected on the basis of known elements of the intelligent network.

Smyk does not deal with a method of <u>routing in an IP data network</u>.

The claimed invention enables more than one session to be maintained at the same time with more than one target, and the routing can be adapted during the session and without interrupting the session to any current requirements or preferences. That is, even if the calling party and the called party (the destination address) do not change, the routing may still be changed. See, for example, Fig. 3, which illustrates a change from connection 6 to connection 7.

Once more in summary, Voit and Smyk deal with VoIP communication. As such, they are only marginally related with the current invention that deals with Internet Protocol data network communication. In a VoIP transmission, the calling user opens a call which is routed once through the network (via an ITG). If the user wants to change the routing, the call must be interrupted. There is no possibility to reroute without interrupting and then reestablishing the call.

The invention, of course, enables establishing an initial route (the routing is not predetermined in accordance with a registered provider, quality, cost, etc.) and it can be changed at any time during the “call” without interrupting the connection. See, for instance, the introductory text and the description of Fig. 3 of the instant application.

It is furthermore an important aspect of VoIP telephony that a plurality of networks must be involved, namely, the Internet or an intranet, and a PSTN (Voit: the receiving side, Smyk: the calling side). Much in contrast, the claimed invention renders the makeup of the networks entirely unimportant.

When judging the teachings of the references, one must not loose sight of the fact that Smyk hails from 1997. At that time, the technology for rendering an intelligent network IN – which pertains to telephone voice transmission – was used (and suitable) only for PSTN systems. The first steps towards adapting IN features for mobile radio communications networks (standardization, CAMEL) only started at approximately that time. Intelligent networks INs for packet-switched networks (Internet, intranet) are <u>not</u> known to date.

The further reference Dobbins et al. (US 6,147,995, "Dobbins") was cited with regard to claims 5 and 6. These claims deal with incorrect or unknown destination addresses in the data packet. While the teaching of Dobbins and its contribution to "address correction" in switched networks is appreciated, it is indeed irrelevant with regard to the primary references Voit and Smyk. There, an incorrect or incomplete telephone number (i.e., the destination address) cannot be corrected. Such a call attempt will simply fail. On the other hand, the address to the destination ITG which handles the called telephone terminal in the PSTN, is <u>not</u> contained in the packet received from the calling user. Instead, the network node determines the address of the pertinent ITG and transmits the packet to that ITG. Again, the destination ITG is determined based on the <u>called user</u>.

Srinivasan (US 6,145,002) was cited with regard to the "helpdesk" claims. We acknowledge the teaching of the secondary reference. The teaching does not in any way modify Voit and Smyk with regard to the base claims and it cannot cure the shortcomings of the basic combination. That is, the independent claims are clearly patentable over the combined teachings of Voit, Smyk and Srinivasan.

It is strongly believed that the claims are now in condition for allowance. Should the Examiner disagree and/or should any further formalistic objections remain, counsel herewith requests an

Interview

with the Examiner. The Examiner is requested to telephone counsel should the claims not yet be considered allowable.

Petition for extension is herewith made. Counsel's payment in the amount of $1020.00 for an extension of three months is enclosed. In view of the foregoing, reconsideration and allowance of claims 1 and 3-37 are solicited.

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